User's Guide

Hammerfall® DSP System

MADI

TotalMix™

SyncAlign® ZLM® SyncCheck®

SteadyClock™

PCI Busmaster Digital I/O Card
64 Channels MADI Interface
24 Bit / 192 kHz Digital Audio
Stereo Analog Monitoring
64 x 64 Matrix Router
2 x MIDI I/O
MIDI embedded in MADI
Quick Boot

Contents

1 Introduction 4
2 Package Contents 4
3 System Requirements 4
4 Brief Description and Characteristics 4
5 Technical Specifications
5.1 Digital 5
5.2 Inputs 5
5.3 Outputs 6
5.4 Analog 6
5.5 Transfer Modes: Resolution/Bits per Sample 7
6 Hardware Installation 7
7 Driver Installation
7.1 Windows 2000/XP 8
7.2 Driver Update 8
7.3 Flash Update 8
7.4 Deinstalling the Drivers 9
7.5 Linux/Unix 9
8 Connectors
8.1 Overview 10
8.2 MADI I/Os 10
8.3 Word Clock I/O 11
8.4 Analog Output 12
8.5 MIDI 12
8.6 Internal Jumper 12
9 Operation and Usage
9.1 Playback 13
9.2 DVD Playback (AC-3 / Multichannel) 14
9.3 Low Latency under MME 15
9.4 Multi-Client Operation 15
9.5 Record 16
10 Configuring the HDSP MADI
10.1 General Information 17
10.2 Clock Modes – Synchronization 19
11 Word Clock
11.1 Technical Description and Usage 20
11.2 Cables and Termination 21
11.3 General Operation 21
12 Using more than one Hammerfall DSP 21
13 Operation under ASIO 2.0
13.1 General 22
13.2 Known Problems 22
14 Operation under GSIF 23
15 TotalMix: Routing and Monitoring 24
15.1 Elements of the Surface 25
15.2 Tour de TotalMix 26
15.3 Submix View 27
15.4 Mute and Solo 27
15.5 Hotkeys 27
15.6 Quick Access Panel 28
15.7 Presets 29
15.8 Monitor 30
15.9 Menu Options 30
15.10 Level Meter 31

16 The Matrix
16.1 Elements of the Surface 32
16.2 Usage 32
16.3 Advantages of the Matrix 33
17 TotalMix Super-Features
17.1 ASIO Direct Monitoring 33
17.2 Selection and Group-based Operation 34
17.3 Copy Routings to other Channels 34
17.4 Delete Routings 34
18 MADI Basics 35
19 SteadyClock 36
20 Hotline - Troubleshooting
20.1 General 36
20.2 Installation 37
21 HDSP Software
21.1 DIGICheck 4.0 38
21.2 HDSP Meter Bridge 2.0 38
22 Accessories 39
23 TECH INFO 39
24 Warranty 40
25 Appendix 40
26 CE / FCC Compliance 41

1. Introduction

Thank you for choosing the Hammerfall DSP MADI. This unique audio system is capable of transferring digital audio data directly into a computer, from any device equipped with a MADI interface. Installation is simple, even for the inexperienced user, thanks to the latest Plug and Play technology. The numerous unique features and well thought-out configuration dialog puts the Hammerfall DSP MADI at the very top of the range of digital audio interface cards.

The package includes drivers for Windows 2000/XP. An ALSA driver for Linux is under development (see chapter 7.5).

Our high-performance philosophy guarantees maximum system performance by executing all functions directly in hardware and not in the driver (i.e. the CPU).

2. Package Contents

Please check that your Hammerfall DSP MADI package contains each of the following:

- HDSP MADI PCI card
- HDSP MADI expansion board
- Quick Info guide
- RME Driver CD
- MIDI breakout cable
- Expansion Board ribbon cable (14-conductor)

3. System Requirements

- Windows 2000/XP, Linux
- PCI Interface: a free PCI rev. 2.1 Busmaster slot

Note: Examples and detailed descriptions of suitable audio desktop systems can be found in the Tech Info *RME Reference PCs: Hardware recommendations*.

4. Brief Description and Characteristics

- Hammerfall design: 0% (zero!) CPU load, even using all 128 ASIO channels
- All settings can be changed in real-time
- 8 available buffer sizes/latencies: 1.5 / 3 / 6 / 12 / 23 / 46 / 93 / 186 ms
- 32 channels 96 kHz/24 bit record/playback
- Automatic and intelligent master/slave clock control
- Word clock input and output
- Zero Latency Monitoring: Hardware bypass per track, controlled by Punch in/out
- Enhanced ZLM for latency-free submixes and perfect ASIO Direct Monitoring
- SyncAlign guarantees sample aligned and never swapping channels
- SyncCheck tests and reports the synchronization status of input signals
- 2 x MIDI I/O, 32 channels high-speed MIDI
- DIGICheck DSP: Level meter in hardware, peak- and RMS calculation
- TotalMix: 8192 channel mixer with 40 bit internal resolution
- SteadyClock: Jitter-immune, super-stable digital clock
- Quick Boot technology for immediate loading of the hardware settings

5. Technical Specifications

5.1 Digital

General

- Clocks: Internal, MADI In, Word Clock In
- Low jitter design: < 1 ns in PLL mode, all inputs
- Internal clock: 800 ps jitter, random spread spectrum
- Jitter suppression of external clocks: about 30 dB (2.4 kHz)
- Effective clock jitter influence on AD and DA conversion: near zero
- Provides nearly jitter-free word clock directly from the MADI input signal
- Input PLL ensures zero dropout, even at more than 100 ns jitter
- Sample frequencies: 32 / 44.1 / 48 / 88.2 / 96 / 176.4* / 192* kHz

*Not yet supported

MIDI

- 2 x MIDI I/O via breakout cable
- 1 x MIDI I/O via MADI
- PCI bus based hi-speed operation
- Seperate 128 byte FIFO for input and output
- MIDI state machine in hardware for reduced interrupt request load
- Invisible transmission via User bit of channel 56 (up to 48 kHz)
- Invisible transmission via User bit of channel 28 (up to 96 kHz)
- Invisible transmission via User bit of channel 14 (up to 192 kHz)

5.2 Inputs

MADI

- Coaxial via BNC, 75 Ohm, according to AES10-1991
- High-sensitivity input stage (< 0.2 Vpp)
- Optical via FDDI duplex SC connector
- 62.5/125 and 50/125 compatible
- Accepts 56 channel and 64 channel mode, plus 96k and 192k Frame
- Single Wire: up to 64 channels 24 bit 48 kHz
- Double Wire: up to 32 channels 24 bit 96 kHz
- Quad Wire: up to 16 channels 24 bit 192 kHz
- Lock range: 25 kHz – 54 kHz
- Jitter when synced to input signal: < 1 ns

Word Clock

- BNC, not terminated (10 kOhm)
- Switch for internal termination 75 Ohm
- Automatic Double Speed detection and internal conversion to Single Speed
- SteadyClock guarantees super low jitter synchronization even in varispeed operation
- AC-coupling, not effected by DC-offsets within the network
- Signal Adaptation Circuit: signal refresh through auto-center and hysteresis
- Overvoltage protection
- Level range: 1.0 Vss – 5.6 Vpp
- Lock range: 27 kHz – 200 kHz
- Jitter when synced to input signal: < 1 ns

MIDI

- 32 channels MIDI
- 5-pin DIN jacks
- Optocoupled, ground-free input
- 1 virtual MIDI port (MADI)

5.3 Outputs

MADI

- Coaxial via BNC, 75 Ohm, according to AES10-1991
- Output voltage 600 mVpp
- Output voltage 1.2 Vpp optional, via jumper
- Cable length: more than 100 m
- Optical via FDDI duplex SC connector
- 62.5/125 and 50/125 compatible
- Cable length: more than 500 m
- Generates 56 channel and 64 channel mode, plus 96k and 192k Frame*
- Single Wire: up to 64 channels 24 bit 48 kHz
- Double Wire / 96K Frame: up to 32 channels 24 bit 96 kHz
- 192K Frame: up to 16 channels 24 bit 192 kHz*

*Not yet supported

Word Clock

- BNC
- Max. output voltage: 5 Vpp
- Output voltage @ 75 Ohm: 4.0 Vpp
- Impedance: 10 Ohm
- Frequency range: 27 kHz – 56 kHz

MIDI

- 32 channels MIDI
- 5-pin DIN jacks
- 1 virtual MIDI port (MADI)

5.4 Analog

DA - Stereo Monitor Output (Phones)

- Signal to Noise ratio (SNR): 110 dB RMS unweighted, 112 dBA @ 44.1 kHz (unmuted)
- THD: < - 100 dB, < 0.001 %
- THD+N: < -98 dB, < 0.0015 %
- Crosstalk: > 100 dB
- Frequency response @ 44.1 kHz, -0.5 dB: 1 Hz – 21.1 kHz
- Frequency response @ 96 kHz, -0.5 dB: 1 Hz – 43.5 kHz
- Frequency response @ 192 kHz, -0.5 dB: 1 Hz – 70 kHz
- Output: 6.3 mm TRS jack
- Output impedance: 50 Ohm
- Output level @ 0 dBFS: +13 dBu

5.5 Transfer Modes: Resolution / Bits per Sample

ASIO:
- 24 or 32 bit, 4 byte (stereo 8 byte)

This format is compatible with 16-bit and 20-bit. Resolutions below 24-bit are handled by the audio application.

MME:
- 16 bit, 2 byte (stereo 4 byte)
- 20 bit, 3 byte MSB (stereo 6 byte)
- 20 bit, 4 byte MSB (stereo 8 byte)
- 24 bit, 3 byte (stereo 6 byte)
- 24 bit, 4 byte MSB (stereo 8 byte)
- 32 bit, 4 byte (stereo 8 byte)

The card supports multi-device operation only, channel interleave operation is not supported.

6. Hardware Installation

Before installing the PCI card, please make sure the computer is switched off and the power cable is disconnected from the mains supply. Inserting or removing a PCI card while the computer is in operation will cause irreparable damage to both motherboard and card!

1. Disconnect the power cord and all other cables from the computer.
2. Remove the computer's housing. Further information on how to do this can be obtained from your computer´s instruction manual.
3. Important: Before removing the HDSP MADI from its protective bag, discharge any static in your body by touching the metal chassis of the PC.
4. Connect the HDSP MADI card with the Expansion Board using the supplied flat ribbon cable. Note: The connectors on the cable cannot be plugged in the wrong way round.
5. Insert the HDSP MADI firmly into a free PCI slot, press and fasten the screw.
6. Insert the Expansion Board and fasten the screw.
7. Replace the computer's housing.
8. Reconnect all cables including the power cord.

Note: If neither word clock I/O nor MIDI I/O is required, it is not necessary to install the Expansion Board at all (i.e. leave out steps 4 and 6).

7. Driver Installation

7.1 Windows 2000/XP

After the PCI card has been installed correctly (see 6. Hardware Installation), and the computer has been switched on, Windows will recognize the new hardware component and start its ‘Hardware Wizard’. Insert the RME Driver CD into your CD-ROM drive, and follow further instructions which appear on your computer screen. The driver files are located in the directory **\HDSP_MADI_w2k** on the RME Driver CD.

Windows will install the Hammerfall DSP System driver, and will register the card in the system as a new audio device. After a reboot the HDSP MADI is ready for use.

HDSP MADI can be easily configured using the HDSP Settings dialog (see section 10.1)

In case the warning messages 'Digital signature not found', 'Do not install driver', 'not certified driver' or similar come up: simply ignore them and continue with the installation.

In case the Hardware Wizard does not show up automatically after installation of the card, do not attempt to install the drivers manually! An installation of drivers for non-recognized hardware will cause a blue screen when booting Windows!

7.2 Driver Update

RME's driver updates often include a new madi.inf file. Also the revision number of the hardware might change (after a flash update). To prevent Windows 2000/XP from using an old madi.inf, or to copy some of the old driver files, be sure NOT to let Windows search for the driver! Instead tell Windows what to do.

Under Control Panel /System /Device Manager /Sound, Video and Game Controllers /RME Hammerfall DSP MADI/Properties /Driver you'll find the 'Update Driver' button. Select '**Install from a list or specific location (advanced)**', click '**Next**', select '**Don't search I will choose the driver to install**', click '**Next**', then '**Have Disk**'. Now point to the driver update's directory.

7.3 Flash Update

The Flash Update Tool updates the HDSP MADI's hardware to the latest version. It requires an already installed driver.

Start the program fut_madi_xxx.exe. The Flash Update Tool displays the current version of the HDSP MADI (200 or up), and whether it needs an update or not. If so, then simply press the 'Update' button. A progress bar shows how several actions are performed. When the flash update process is finished, 'Success' will be displayed.

If more than one interface cards are installed, all cards can be flashed by changing to the next tab and repeating the process.

After the update the PCI card needs to be resettet. This is done by powering down and shutting off the PC. A warm boot is not enough.

When the update fails (status: failure), the card's second BIOS will be used from the next cold boot on (Secure BIOS Technology). Therefore the card stays fully functional. The flash process should then be tried again on a different computer.

Note for Windows 2000/XP users: Because of the changed hardware revision, Windows 2000/XP will start the hardware assistant and wants to install new drivers. Do NOT let Windows search for new drivers, but follow the instructions given in chapter 7.2 and manually perform a driver update.

7.4 Deinstalling the Drivers

A deinstallation of the HDSP's driver files is not necessary – and not supported by Windows anyway. Thanks to full Plug & Play support, the driver files will not be loaded after the hardware has been removed. If desired these files can then be deleted manually.

Unfortunately Windows Plug & Play methods do not cover the additonal autorun entries of TotalMix, the Settings dialog, and the registering of the ASIO driver. Those entries can be removed from the registry through a software deinstallation request. This request can be found (like all deinstallation entries) in *Control Panel, Software*. Click on the entry 'RME Hammerfall DSP MADI'.

7.5 Linux/Unix

An ALSA driver for Linux/Unix is already under development and should be available soon. Even TotalMix has been ported to Linux. Further information on ALSA is available at

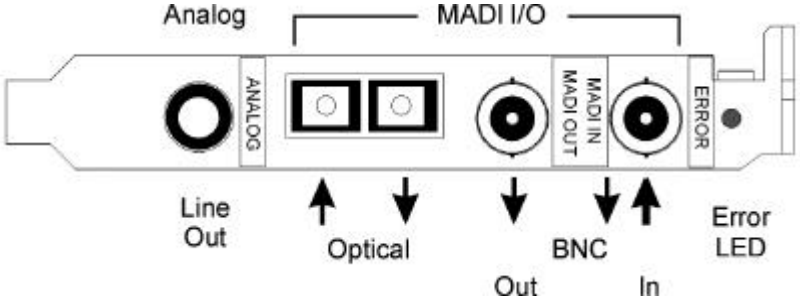
http://www.alsa-project.org

8. Connectors

8.1 Overview

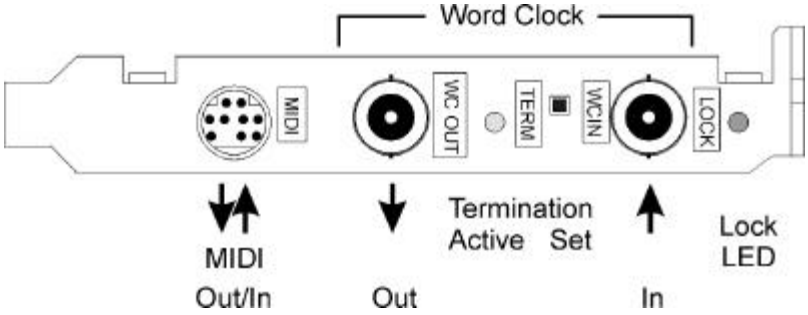
HDSP MADI consists of the main PCI board and an Expansion Board. All the essential electronics are located on the PCI card, so it will also work without the Expansion Board.

The main board's bracket has two MADI interfaces, optical and coaxial input and output each, a MADI erro LED and the analog line/headphone output. Configuration of inputs and outputs is done via the Settings dialog (started by clicking on the hammer symbol in the system tray).



Identical signals are available at both the optical and the coaxial output. An obvious use for this would be to simply connect two devices, i.e. using the HDSP MADI as a splitter.

The Expansion Board's bracket has the word clock input and output. Next to the input BNC socket, a green LED displays the word clock input's lock status. Between the BNC sockets, word clock termination can be activated and verified by a yellow LED.



The included breakout cable is connected to the 9-pin Mini-DIN connector and provides two MIDI inputs and outputs.

8.2 MADI I/Os

The BNC input's ground-free design is built according to AES10-1991. The input impedance is 75 Ohm. It will operate error-free from about 180 mVpp.

The optical input and output uses a FDDI (ISO/IEC 9413-3) compatible optical module, according to AES10-1991. More information can be found in chapter 18, MADI Basics.

HDSP MADI includes automatic input selection (Safe Mode Input). In case the current input signal fails, the unit switches to the other input immediately. This mode, called redundancy mode, offers improved safety against errors on the transmission line. Switching the inputs is done in about one second. Redundancy operation is disaplayed in the Settings dialog.

The BNC output is built according to AES10-1991. The output's impedance is 75 Ohm. The output voltage will be 600 mVpp when terminated with 75 Ohm. Changing the internal blue jumper X4 to the upper position, the output voltage is increased to 1.2 Vpp. This setting is not intended to be used in normal operation. But in case of a very long or 'lossy' coaxial cable, this setting may ensure an error-free operation of the transmission line.

8.3 Word Clock I/O

HDSP MADI includes SteadyClock, guaranteeing an excellent performance in all clock modes. Its highly efficient jitter suppression refreshes and cleans up any clock signal, and provides it as reference clock at the BNC output (see section 19).

Input

The transformer-isolated word clock input is loacated on the Expansion Board. It is activated via *Pref. Sync Ref* / *Wordclock*. As soon as a valid word clock signal is detected, the green 'Lock' LED beside the input jack lights up, and in the Settings dialog the field *AutoSync Ref* shows *Word*. Below the detected frequency of the word clock is shown.

The word clock input is shipped as high impedance type (not terminated). A push switch allows to activate internal termination (75 Ohms). The switch is found between the BNC jacks. Use a small pencil or similar and carefully push the blue switch so that it snaps into its lock position. When termination is active the yellow TERM LED will be lit. Another push will release it again and de-activate the termination.

Due to the HDSP MADI's outstanding clock control a synchronization of the output signal to the input signal is not only possible at identical sample rates, but also at half, quarter, double and quad sample rates.

Example 1: A playback or recording at 44.1 kHz can be synchronized via an external signal of 44.1 kHz, 88.2 kHz or 176.4 kHz.

Example 2: A playback or recording at 192 kHz can be synchronized via an external signal of 48 kHz, 96 kHz or 192 kHz.

The input accepts all those frequencies fully automatically.

Thanks to RME's *Signal Adaptation Circuit*, the word clock input still works correctly even with heavily mis-shaped, dc-prone, too small or overshoot-prone signals. Thanks to automatic signal centering, 300 mV (0.3V) input level are sufficient in principle. An additional hysteresis reduces sensitivity to 1.0 V, so that over- and undershoots and high frequency disturbances don't cause a wrong trigger.

Output

The word clock output of the HDSP MADI is always active. It provides the current sample frequency of the HDSP MADI as word clock signal. As long as the HDSP MADI operates in 'Master' mode (field 'Clock Mode'), the word clock will be fixed to the current sample rate. In 'AutoSync' mode the sample rate is identical to the one present at the currently chosen input (Pref. Sync Ref). Without a valid input signal, the card will change between the inputs automatically. As long as no valid input signal is found, the card will stay in Master mode. This way, the card will always generate a valid output signal.

The word clock signal received by the HDSP MADI can be distributed to other devices by using the word clock output. With this the usual T-adapter can be avoided, and the HDSP MADI operates as *Signal Refresher*. This kind of operation is highly recommended, because

- Input and output are phase-locked and in phase (0°) to each other
- SteadyClock removes nearly all jitter from the input signal
- the exceptional input (1 Vpp sensitivity instead of the usual 2.5 Vpp, dc cut, Signal Adaptation Circuit) plus SteadyClock guarantee a secure function also with most critical word clock signals.

8.4 Analog Output

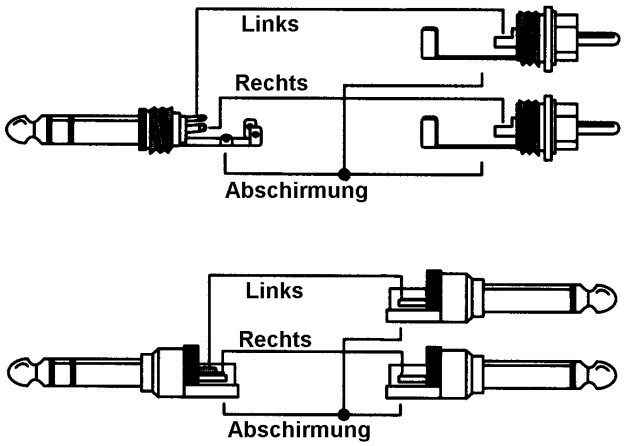
HDSP MADI offers a hi-quality analog monitor output. The short circuit protected stereo line output provides high output level, low impedance, and is available via a 6.3 mm (1/4") TRS jack. Therefore it is also suitable for a direct use with headphones.

The analog output is directly driven from the channels 63/64. Its output volume is controlled by the hardware output faders of channel 63/64 in TotalMix. Additionally the analog output can play back any input or playback signal (submix, for example Preset 5, 6 and 7).

RME's unique Speaker Protection reduces noise when switching the computer on and off, so there is no problem even when using active monitors.

In case the output should operate as line out, an adapter TRS plug to RCA phono plugs, or TRS plug to TS plugs is required.

The pin assignment follows international standards. The left channel is connected to the tip, the right channel to the ring of the TRS jack/plug.



8.5 MIDI

Hammerfall DSP MADI offers two MIDI I/Os via 5-pin DIN jacks. The MIDI ports are added to the system by the driver. Using MIDI capable software, these ports can be accessed as MADI MIDI In 1 (1), MADI MIDI In 2 (2), MADI MIDI Out 1 (1) and MADI MIDI Out 2 (2). The brackets include the card number.

The MIDI inputs can not operate multiclient, so a MIDI input signal can not be distributed to several programs at the same time. Such a functionality can be achieved with third party tools.

The third MIDI port, MADI MIDI In 3 (1) and MADI MIDI Out 3 (1), receives and transmits MIDI data via MADI. This allows for a direct communication between systems with HDSP MADI cards. Additionally MIDI data can be transmitted from/to RME's ADI-648. Also the ADI-648 can be MIDI remote controlled without any additional line or cabling between computer (MADI card) and ADI-648.

8.6 Internal Jumper

The internal blue jumper X4 is neither input nor output. It allows to change the voltage level at the coaxial MADI output. In the lower position, the card generates 600 mVpp (according to the specification). in case of a very long or 'lossy' coaxial cable, a higher output level might help to maintain an error-free operation. In the upper position, the output level is 1.2 Vpp.

9. Operation and Usage

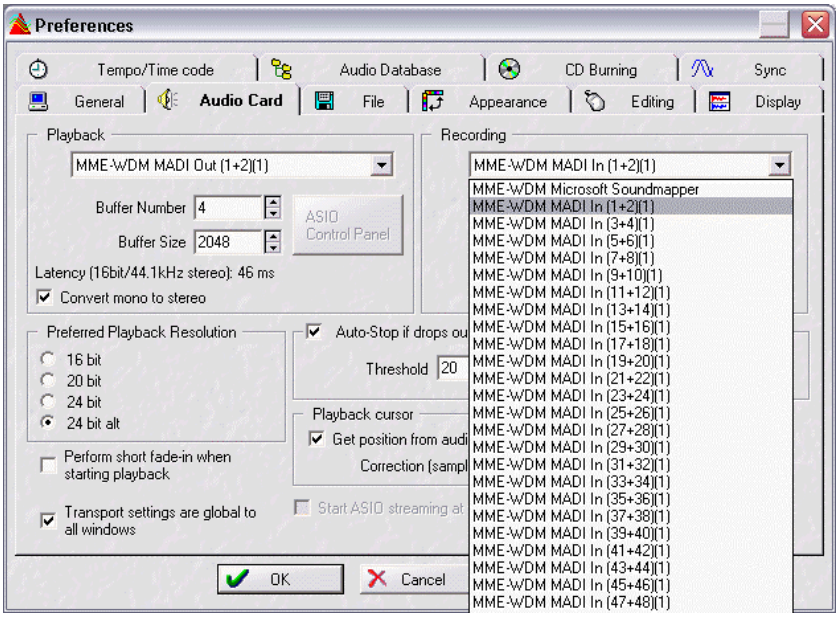
9.1 Playback

The HDSP system can play back audio data in supported formats only (sample rate, bit resolution). Otherwise an error message appears (for example at 22 kHz and 8 bit).

In the audio application being used, HDSP must be selected as output device. This can often be found in the *Options, Preferences* or *Settings* menus under *Playback Device, Audio Devices, Audio* etc. We recommend using 24-bit resolution for playback, to make full use of the HDSP's potential.

We strongly recommend switching all system sounds off (via >Control Panel /Sounds<). Also HDSP should not be the *Preferred Device* for playback, as this could cause loss of synchronization and unwanted noises. If you feel you cannot do without system sounds, you should consider buying a cheap Blaster clone and select this as *Preferred Device* in >Control Panel /Multimedia /Audio<.

The RME Driver CD includes step by step instructions for configuring many popular audio applications, found in the directory **\rmeaudio.web\english\techinfo\conf**.



The screenshot to the right shows a typical configuration dialog as displayed by a (stereo) wave editor. After selecting one of the 32 playback devices, audio data is sent to the according audio channels.

Increasing the number and/or size of audio buffers may prevent the audio signal from breaking up, but also increases latency i.e. output is delayed. For synchronized playback of audio and MIDI (or similar), be sure to activate the checkbox 'Get position from audio driver'. Even at higher buffer settings in a mixed Audio/MIDI environment, sync problems will not arise because the Hammerfall DSP always reports the current play position correctly (even while recording - essential for chase lock synchronization).

9.2 DVD-Playback (AC-3 / DTS / Multichannel) under MME

AC-3 / DTS
When using popular DVD software player like WinDVD and PowerDVD, their audio data stream can be sent to any AC-3/DTS capable receiver using the HDSP MADI. For this to work, in most cases a device of the HDSP MADI has to be selected in 'Control Panel/Sounds and Multimedia/Audio'. Also check 'use preferred device only'.

You will notice that the DVD software's audio properties now allow to use 'SPDIF Out' or to 'activate SPDIF output'. When selecting these, the software will transfer the non-decoded digital multichannel data stream to the Hammerfall DSP. Naturally a successful decoding also requires a MADI to AES converter, converting the playback channels to AES or SPDIF.

Note: AC-3 sounds like chopped noise at highest level.

Multichannel
PowerDVD and WinDVD can also operate as software decoder, sending a DVD's multichannel data stream directly to any channels of the HDSP MADI. Supported are all modes, from 2 to 8 channels, at 16 bit resolution and 48 kHz sample rate.

For this to work an output wave device of the HDSP has to be selected in 'Control Panel/Sounds and Multimedia/Audio'. Also check 'use preferred device only'. PowerDVD's audio properties now lists several multichannel modes. If one of these is selected, PowerDVD plays back the decoded analog multichannel data via the HDSP MADI.

The device selected as *Preferred Playback Device* defines the first playback channel. Choosing ADAT 3/4 and 6-channel mode, playback will happen on channels 3 to 8. Choosing MADI 3/4 and 6-channel mode, playback will happen on channels 3 to 8.

The channel assignment using PowerDVD is:

1 (first chosen playback channel) - Left
2 - Right
3 - Center
4 - LFE (Low Frequency Effects)
5 - SR (Surround Right)
6 - SL (Surround Left)

Note 1: Setting the card to be used as system playback device is against common sense, as professional cards are not specialized to play back system sounds, and shouldn't be disturbed by system events. To prevent this, be sure to re-assign this setting after usage, or to disable any system sounds (tab Sounds, scheme 'No audio').

Note 2: The DVD player will be synced backwards from the HDSP card. This means when using AutoSync and/or word clock, the playback speed and pitch follows the incoming clock and sample rate.

Note 3: PowerDVD 5 no longer supports a starting channel other than channel 1.

Note 4: In WinDVD 5, a channel-seperated 5.1 playback using the HDSP requires a change in the registry. Start regedit, go to *HKEY_CURRENT_USER* / *Software* / *InterVideo* / *DVD5* / *AUDIOCHAN*, and set its value to *4* (hexadecimal). Additionally select 'Waveout' as Audio Renderer in the Audio configuration dialog. **Warning: Changes to the registry are done at your own risk. Danger of complete data loss!**

9.3 Low Latency under MME (Buffer Size Adjustment)

Using Windows 95 or 98 the MME buffer size was nothing to worry about. Latencies below 46 ms were not possible. Meanwhile both computers and operating system have become much more powerful, and with Windows 2000/XP latencies far lower can be used. SAWStudio and Sonar allowed to use such low settings from the start. Sequoia was updated in version 5.91, WaveLab in version 3.04.

In the HDSP's Settings dialog the MME buffersize (in fact the DMA buffer size) is set with the same buttons as the ASIO buffer size. Our test computers allow to use settings down to 64 samples without clicks. Please note that this setting only defines the buffer size of the hardware. The true and effective latency is configured within the MME application!

Attention: the DMA buffers must not be larger than the application's buffers. This case can happen unnoticed when using ASIO and MME at the same time (multi-client) and setting ASIO to 186 ms, while the buffers in the MME application are still set for a lower latency. Playback will be stuttering and audio will be distorted.

Example: when you set the Hammerfall to 512 you can't use 128 in any program. But setting DMA to 128 allows to use 128 and all higher values within the software.

Please also note that this is a 'you're welcome to try' feature. We can't guarantee that you will be able to use 3 or 6 ms with MME. Simply check out by yourself which lowest setting your system and software allows. Some motherboards with insufficient PCI bandwidth (especially VIA based) suffer from crackling at settings below 512. Be sure to set the buffer size to 512 or higher in such a case (or trash the motherboard…).

9.4 Multi-Client Operation

RME audio cards support multi-client operation. This means several programs can be used at the same time. Also all formats, like ASIO, MME and GSIF can be used simultaneously. The use of multi-client operation requires to follow two simple rules:

- *Multi-client operation requires identical sample rates!*

It is not possible to use one software with 44.1 kHz and the other with 48 kHz.

- *Different software can not use the same channels at the same time.*

If for example Cubase uses channels 1/2 (default in Cubase, Master bus), this playback pair can't be used in Gigasampler/Studio (GSIF) nor under MME (WaveLab etc) anymore (the inputs can be used at the same time). This is no limitation at all, because TotalMix allows any output routing, and with this a playback of multiple software on the same hardware outputs.

ASIO Multi-client
RME audio cards support ASIO multi-client operation. It is possible to use more than one ASIO software at the same time. Again the sample rate has to be identical, and each software has to use its own playback channels. The inputs can be used simultaneously.

An exception is our sophisticated tool *DIGICheck*. It operates like an ASIO host, using a special technique to access playback channels already occupied. Therefore DIGICheck is able to perform an analyzis and display of playback data from any software, no matter which format the software uses.

9.5 Recording

Unlike analog soundcards which produce empty wave files (or noise) when no input signal is present, digital I/O cards always need a valid input signal to start recording.

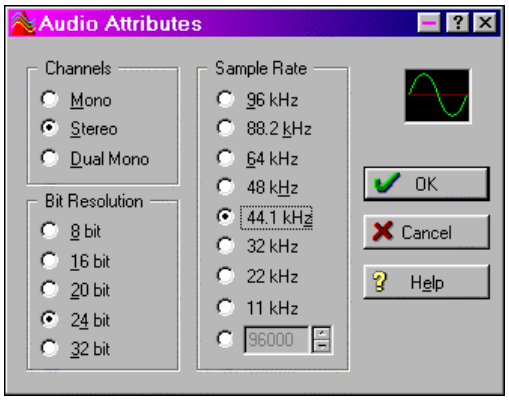
To take this into account, RME has included two unique features in the HDSP MADI: a comprehensive I/O signal status display (showing sample frequency, lock and sync status) in the Settings dialog, and the protective *Safe Mode / Frequency* function.

If a 48 kHz signal is fed to the input and the application is set to 44.1 kHz, recording is prohibited. This prevents faulty takes, which often go unnoticed until later on in the production. Such tracks appear to have the wrong playback rate - the audio quality as such is not affected.

The sample frequency shown in the Settings dialog (see chapter 10, screenshot Settings) is useful as a quick display of the current configuration (the board itself and all connected external equipment). If no sample frequency is recognized, it will read 'No Lock'.

With this configuring any suitable audio application for digital recording is simple. After selecting the required input, Hammerfall DSP displays the current sample frequency. This parameter can then be changed in the application's audio attributes (or similar) dialog.

The screenshot to the right shows a typical dialog used for changing basic parameters such as sample frequency and resolution in an audio application.



Any bit resolution can be selected, providing it is supported by both the audio hardware and the software. Even if the input signal is 24 bit, the application can still be set to record at 16-bit resolution. The lower 8 bits (and therefore any signals about 96dB below maximum level) are lost entirely. On the other hand, there is nothing to gain from recording a 16-bit signal at 24-bit resolution - this would only waste precious space on the hard disk.

It often makes sense to monitor the input signal or send it directly to the output. This can be done at zero latency using **TotalMix** (see chapter 15).

TotalMix also includes a useful automatic real-time monitor function, see chapter 15.8 for details. Activating record in the application causes the input signal to be routed according to the current mixer settings.

Currently two solutions exist which enable an automated control of real-time monitoring. ZLM (Zero Latency Monitoring) allows monitoring in Punch I/O mode - with this the system behaves like a tape machine. This method has been implemented in all versions of Samplitude (by SEK'D), and can be activated using the global track option 'Hardware monitoring during Punch'.

The other solution is Steinberg's ASIO protocol with our ASIO 2.0 drivers and all ASIO 2.0 compatible programs. When 'ASIO Direct Monitoring' has been switched on the input signal is routed in real-time to the output whenever Record is started.

10. Configuring the HDSP MADI

10.1 General Information

Configuring the HDSP system is done using its own settings dialog. The panel 'Settings' can be opened in two different ways:

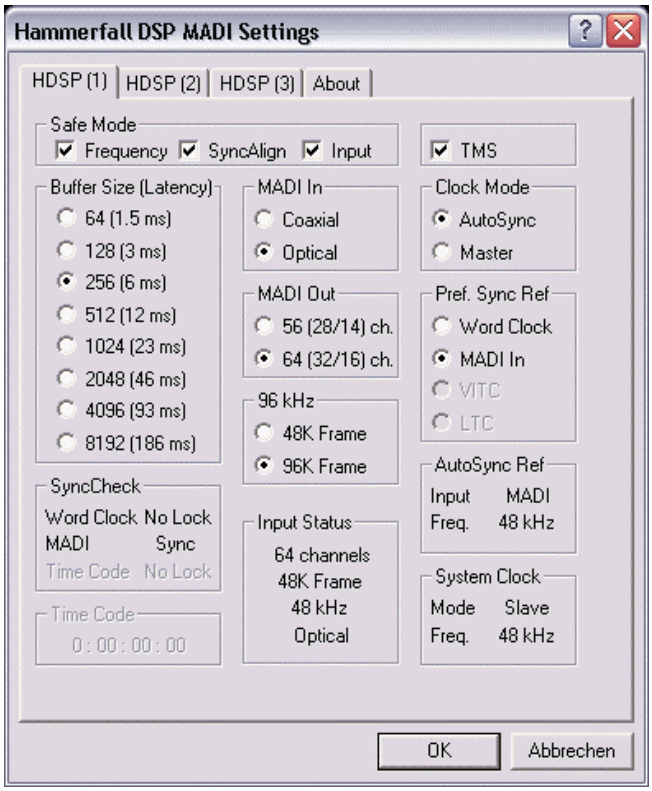
- by clicking on the hammer icon in the Taskbar's system tray

The mixer of the Hammerfall DSP System (TotalMix) can be opened in two different ways:

- by clicking on the mixer icon in the Taskbar's system tray

The Hammerfall DSP's hardware offers a number of helpful, well thought-of practical functions and options which affect how the card operates - it can be configured to suit many different requirements. The following is available in the 'Settings' dialog:

- Input selection
- Output mode
- Synchronization behaviour
- Input and output status display
- Time code display*



Any changes made in the Settings dialog are applied immediately - confirmation (e.g. by clicking on OK or exiting the dialog) is not required. However, settings should not be changed during playback or record if it can be avoided, as this can cause unwanted noises. Also, please note that even in 'Stop' mode, several programs keep the recording and playback devices open, which means that any new settings might not be applied immediately.

The status displays at the bottom of the dialog box give the user precise information about the current status of the system, and the status of all signals. 'SyncCheck' indicates whether there is a valid signal for each input ('Lock' or 'No Lock'), or if there is a valid *and* synchronous signal ('Sync'). The 'AutoSync Ref' display shows the input and frequency of the current sync source.

*'Time Code' displays time information received from the optional Sync Module.

Quick Boot
All the card's settings described below are stored in a hardware memory, and are loaded immediately after a power-on of the computer. In clock mode master even the last used sample rate is set. Directly after switching on the computer, a stable and predictable clock state is found at the HDSP MADI's outputs. This advanced technology completely eliminates disturbing noises and clock network problems during power-up or re-boot times.

Safe Mode
Frequency verifies the current input signal against the settings in the record program. When de-activated a recording will always be allowed, even with non-valid input signals.

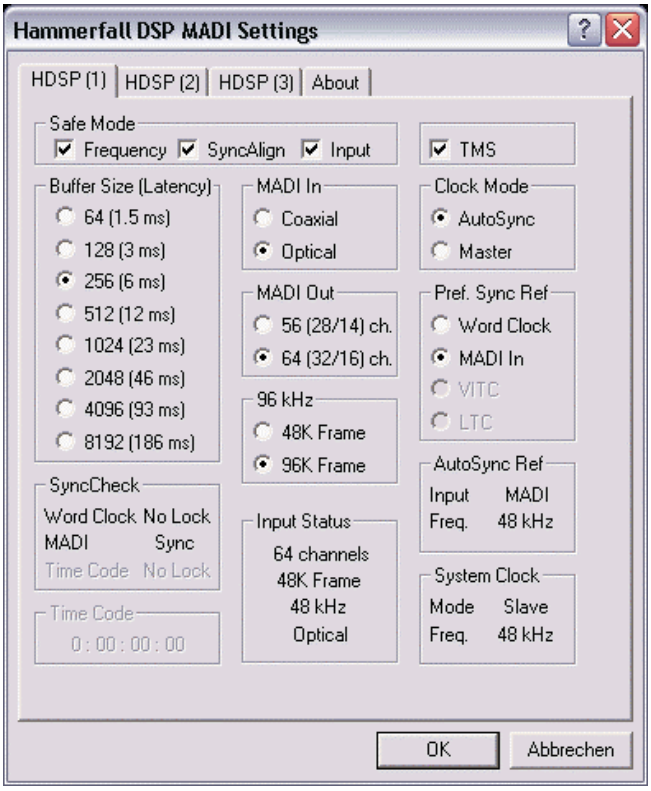
SyncAlign ensures fully sample-aligned channels within MME multitrack software. This option should only be switched off in case the used software does not work at all with SyncAlign active.

Input activates redundancy operation. If the current input signal fails, the other input will be used immediately, provided a valid signal is found there.

TMS activates the transmission of Channel Status data and Track Marker information from the MADI input signal.

Buffer Size
The setting *Buffer Size* determines the latency between incoming and outgoing ASIO and GSIF data, as well as affecting system stability (see chapter 13). Under Windows MME this setting determines the DMA buffer size (see chapter 9.3).

SyncCheck
'SyncCheck' indicates for coaxial and optical MADI input whether there is a valid signal ('Lock' or 'No Lock'), or a valid *and* synchronous signal ('Sync'). The 'AutoSync Ref' display shows the input and frequency of the current sync source.



MADI In
Defines the input for MADI signal. 'Coaxial' relates to the BNC socket, 'Optical' to the optical input.

MADI Out
Defines the format of the MADI output signal. MADI can be a 56 or 64 channel signal. Sample rates higher than 48 kHz can be transmitted using the normal 48K Frame, or using a native 96K Frame at the card's output.

Clock Mode
The card can be configured to use its internal clock (Master), or the clock source pre-defined via *Pref. Sync Ref* (AutoSync).

Pref. Sync Ref
Used to pre-select the desired clock source. If the selected source isn't available the card will change to the other one. The currently used clock source and sample rate is displayed in the *AutoSyncRef* display. The automatic clock selection checks and changes between the clock sources MADI and word clock.

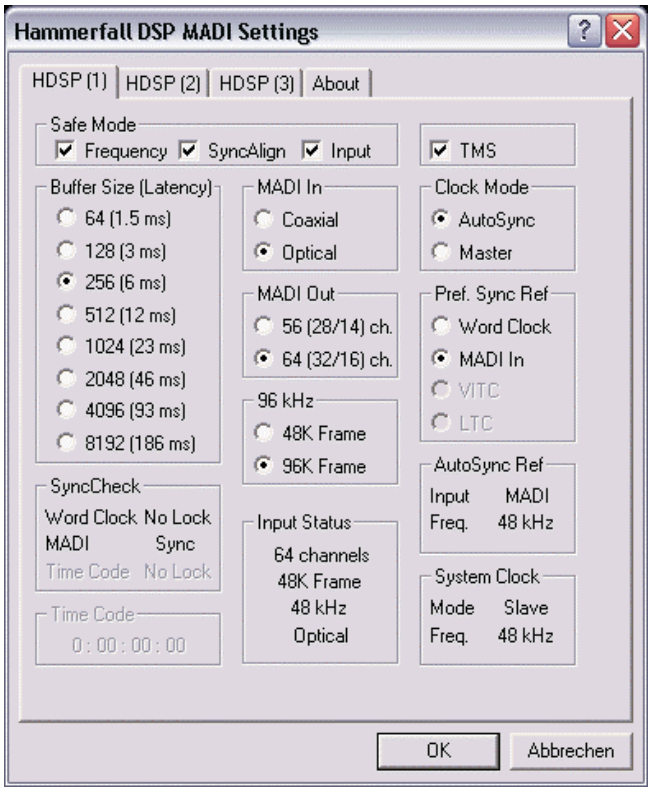
System Clock
Shows the current clock state of the HDSP system. The system is either Master (using its own clock) or Slave (AutoSync Ref).

10.2 Clock Modes - Synchronization

AutoSync
The HDSP MADI has been equipped with *AutoSync*, an automatic clock source selection, which adopts the first available input with a valid digital signal as the clock reference input. The input currently used as sync reference is shown in the *AutoSync Ref* status field, together with its sample frequency.

AutoSync guarantees that normal record and record-while-play will always work correctly. In certain cases however, AutoSync may cause feedback in the digital carrier, so synchronization breaks down. To remedy this, switch the HDSP's clock mode over to 'Master'.

Via *Pref. Sync Ref* (preferred synchronization reference) a preferred input can be defined. As long as the card sees a valid signal there, this input will be designated as the sync source, otherwise the other inputs will be scanned in turn. If none of the inputs are receiving a valid signal, the card automatically switches clock mode to 'Master'.



Thanks to its AutoSync technique and lightning fast PLLs, the HDSP is not only capable of handling standard frequencies, but also any sample rate between 25 and 200 kHz. Even the word clock input, most often used in varispeed operation, allows any frequency between 25 kHz and 200 kHz.

The HDSP MADI's outstanding clock control allows for a synchronization of the output signal to the input signal not only at identical sample rates, but also at half, quarter, double and quad sample rates. A playback of 192 kHz can easily be synchronized via a 48 kHz clock source.

SyncCheck
If several digital devices are to be used simultaneously in a system, they not only have to operate with the same sample frequency but also be synchronous with each other. This is why digital systems always need a single device defined as 'master', which sends the same clock signal to all the other ('slave') devices. RME's exclusive *SyncCheck* technology (first implemented in the Hammerfall) enables an easy to use check and display of the current clock status. The 'SyncCheck' field indicates whether no signal ('No Lock'), a valid signal ('Lock') or a valid *and* synchronous signal ('Sync') is present at each of the digital clock source inputs. The 'AutoSync Ref' display shows the current sync source and the measured frequency.

In practice, SyncCheck provides the user with an easy way of checking whether all digital devices connected to the system are properly configured. With SyncCheck, finally anyone can master this common source of error, previously one of the most complex issues in the digital studio world.

11. Word Clock

11.1 Technical Description and Usage

In the analog domain one can connect any device to another device, a synchronization is not necessary. Digital audio is different. It uses a clock, the sample frequency. The signal can only be processed and transmitted when all participating devices share the same clock. If not, the signal will suffer from wrong samples, distortion, crackle sounds and drop outs.

AES/EBU, SPDIF, ADAT and MADI are self-clocking, an additional word clock connection in principle isn't necessary. But when using more than one device simultaneously problems are likely to happen. For example any self-clocking will not work in a loop cabling, when there is no 'master' (main clock) inside the loop. Additionally the clock of all participating devices has to be synchronous. This is often impossible with devices limited to playback, for example CD players, as these have no SPDIF input, thus can't use the self clocking technique as clock reference.

In a digital studio synchronisation is maintained by connecting all devices to a central sync source. For example the mixing desk works as master and sends a reference signal, the word clock, to all other devices. Of course this will only work as long as all other devices are equipped with a word clock or sync input, thus being able to work as slave (some professional CD players indeed have a word clock input). Then all devices get the same clock and will work in every possible combination with each other.

Remember that a digital system can only have one master! If the HDSP MADI's clock mode is set to 'Master', all other devices must be set to 'Slave'.

But word clock is not only the 'great problem solver', it also has some disadvantages. The word clock is based on a fraction of the really needed clock. For example SPDIF: 44.1 kHz word clock (a simple square wave signal) has to be multiplied by 256 inside the device using a special PLL (to about 11.2 MHz). This signal then replaces the one from the quartz crystal. Big disadvantage: because of the high multiplication factor the reconstructed clock will have great deviations called jitter. The jitter of a word clock is typically 15 times higher as when using a quartz based clock.

The end of these problems should have been the so called Superclock, which uses 256 times the word clock frequency. This equals the internal quartz frequency, so no PLL for multiplying is needed and the clock can be used directly. But reality was different, the Superclock proved to be much more critical than word clock. A square wave signal of 11 MHz distributed to several devices - this simply means to fight with high frequency technology. Reflections, cable quality, capacitive loads - at 44.1 kHz these factors may be ignored, at 11 MHz they are the end of the clock network. Additionally it was found that a PLL not only generates jitter, but also also rejects disturbances. The slow PLL works like a filter for induced and modulated frequencies above several kHz. As the Superclock is used without any filtering such a kind of jitter and noise suppression is missing. No wonder Superclock did not become a commonly accepted standard.

The actual end of these problems is offered by the **SteadyClock** technology of the HDSP MADI. Combining the advantages of modern and fastest digital technology with analog filter techniques, re-gaining a low jitter clock signal of 11 MHz from a slow word clock of 44.1 kHz is no problem anymore. Additionally, jitter on the input signal is highly rejected, so that even in real world usage the re-gained clock signal is of highest quality.

This is especially true when extracting the word clock out of a MADI signal. Caused by the MADI format itself, such a signal will have around 80 (!) ns of jitter, which is reduced to about 1 (!) ns by SteadyClock.

11.2 Cables and Termination

Word clock signals are usually distributed in the form of a network, split with BNC T-adapters and terminated with resistors. We recommend using off-the-shelf BNC cables to connect all devices, as this type of cable is used for most computer networks. You will find all the necessary components (T-adapters, terminators, cables) in most electronics and/or computer stores.

To avoid voltage loss and reflections, both the cable itself and the terminating resistor should have an impedance of 75 Ohm. If the voltage is too low, synchronization will fail. High frequency reflection effects can cause both jitter and sync failure.

Ideally all outputs of word clock delivering devices are designed with very low impedance, but all word clock inputs with high impedance, in order to not weaken the signal on the chain. But there are also negative examples, when the 75 Ohms are built into the device and cannot be switched off. In this case the network load is often 2 x 75 Ohms, and the user is forced to buy a special word clock distributor. Note that such a device is generally recommended for larger studios. Also, 75 Ohm cable is almost impossible to find these days. 50 Ohm cable is standard - this will also work as long as the termination resistors are 75 Ohm.

The HDSP MADI's word clock input can be high-impedance or terminated internally, ensuring maximum flexibility. If termination is necessary (e.g. because HDSP MADI is the last device in the chain), activate the switch TERM between the BNC jacks on the Expansion Board so that the yellow TERM LED lights up.

In case the HDSP MADI resides within a chain of devices receiving word clock, plug a T-adapter into the BNC input jack, and the cable supplying the word clock signal to one end of the adapter, but connect the free end to the next device in the chain via a further BNC cable. The last device in the chain should be terminated using another T-adapter and a terminator plug as described in the previous paragraph. Some devices (like the HDSP MADI) have switchable 75 Ohm resistors, which saves both T-adapter and terminator.

Due to the outstanding SteadyClock technology of the HDSP MADI, we recommend not to pass the input signal via T-adapter, but to use the HDSP MADI's word clock output instead. Thanks to SteadyClock, the input signal will both be freed from jitter and - in case of loss or drop out – be held at the last valid frequency.

11.3 General Operation

The green 'Lock' LED at the Expansion Board will light up as soon as a word clock signal is detected. Selecting 'Word Clock' in the 'Clock Mode' field will switch clock control over to the word clock signal. As soon as there is a valid signal at the BNC jack, 'AutoSync Ref' will display 'Word'. This message has the same meaning as the green 'Lock' LED, but appears on the monitor, i.e. the user can check immediately whether a valid word clock signal is present and is currently being used.

12. Using more than one Hammerfall DSP

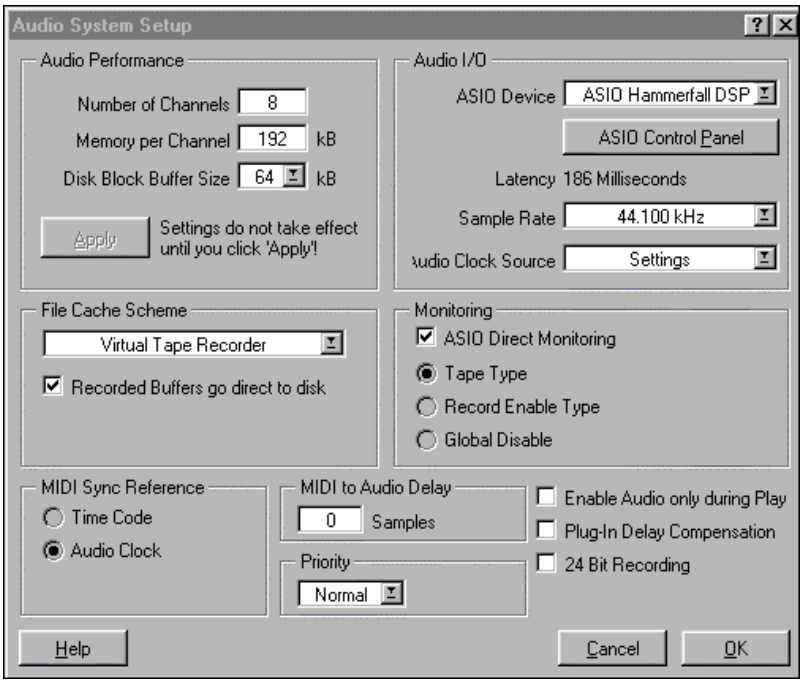
The current drivers support up to three Hammerfall DSP MADI. All cards must be in sync, i.e. have to receive valid sync information (either via word clock or by using AutoSync and feeding synchronized signals).

13. Operation under ASIO 2.0

13.1 General

We will use Steinberg's Cubase VST as an example throughout this chapter. All information provided can easily be adaptated to other programs.

Start the ASIO software and select 'System' from the Audio menu. Select 'ASIO Hammerfall DSP MADI' as the audio I/O device. The 'ASIO system control' button opens the HDSP's Settings dialog (see chapter 10, Configuration).



Hammerfall DSP supports 'ASIO Direct Monitoring' (ADM).

When the sample frequency is set to 88.2 or 96 kHz (Double Speed Mode), the number of available channels is halved.

13.2 Known Problems

In case the used computer has no sufficient CPU-power and/or sufficient PCI-bus transfer rates, then drop outs, crackling and noise will appear. We also recommend to deactivate all PlugIns to verify that these are not the reason for such effects.

Unfortunately some newer UltraATA66 and UltraATA100 hard disk controller (also Raid controller) seem to violate against the PCI specs. To achieve the highest throughput they hog the PCI bus, even in their default setting. Thus when working with low latencies heavy drop outs (clicks) are heard. Try to solve this problem by changing the default setting of the controller (for example by reducing the 'PCI Bus Utilization').

Another common source of trouble is incorrect synchronization. ASIO does not support asynchronous operation, which means that the input and output signals must not only have the same sample frequency, but they must also be in sync. All devices connected to the Hammerfall DSP must be properly configured for Full Duplex operation. As long as SyncCheck (in the Settings dialog) only displays 'Lock' instead of 'Sync', the devices have not been set up properly!

14. Operation under GSIF (Gigasampler Interface)

Windows 2000/XP

The GSIF interface of the HDSP MADI allows direct operation with Gigastudio, with up to 32* channels, 96kHz and 24bit.

Gigastudio requires a lot of the computer's calculation power. An optimum performance is achieved with a stand-alone GSIF PC. However, the GSIF latency is always the same as the one selected for ASIO operation. This can cause performance problems on slower machines when using GSIF and ASIO at the same time.

Please note that the W2k/XP driver fully supports multi-client operation, including the combination MME/ASIO. So for example Cubase, Gigastudio and Sonar can be used simultaneously, provided each of these programs uses its own audio channels exclusively. For example ASIO could use channels 1/2 and Gigastudio (with GSIF) channels 3/4 simultaneously, and so on.

Simultaneous operation of different programs requires to use different channels. For example, if Cubase uses tracks 1/2 these tracks can not be used by Gigastudio.

Common Problems

Please note that Gigastudio is running unexpectedly in the background (thus blocking its assigned audio channels), as soon as the Gigastudio MIDI ports are used – even when Gigastudio itself hasn't been started. This causes a lot of confusion, as the driver seems to behave completely buggy, and the user does not recognize the simple reason for it – for example simultaneous operation of ASIO and GSIF on the same channels.

In case Gigastudio loads well, load gig files too, but won't play at all even when using the virtual keyboard: Go to Hardware/Routing and select a valid MIDI input port. Note that blank is not valid, but <none> is.

*The limitation of 32 channels is caused by Gigastudio 2.54. According to Tascam, Gigastudio 3 will support 64 channels.

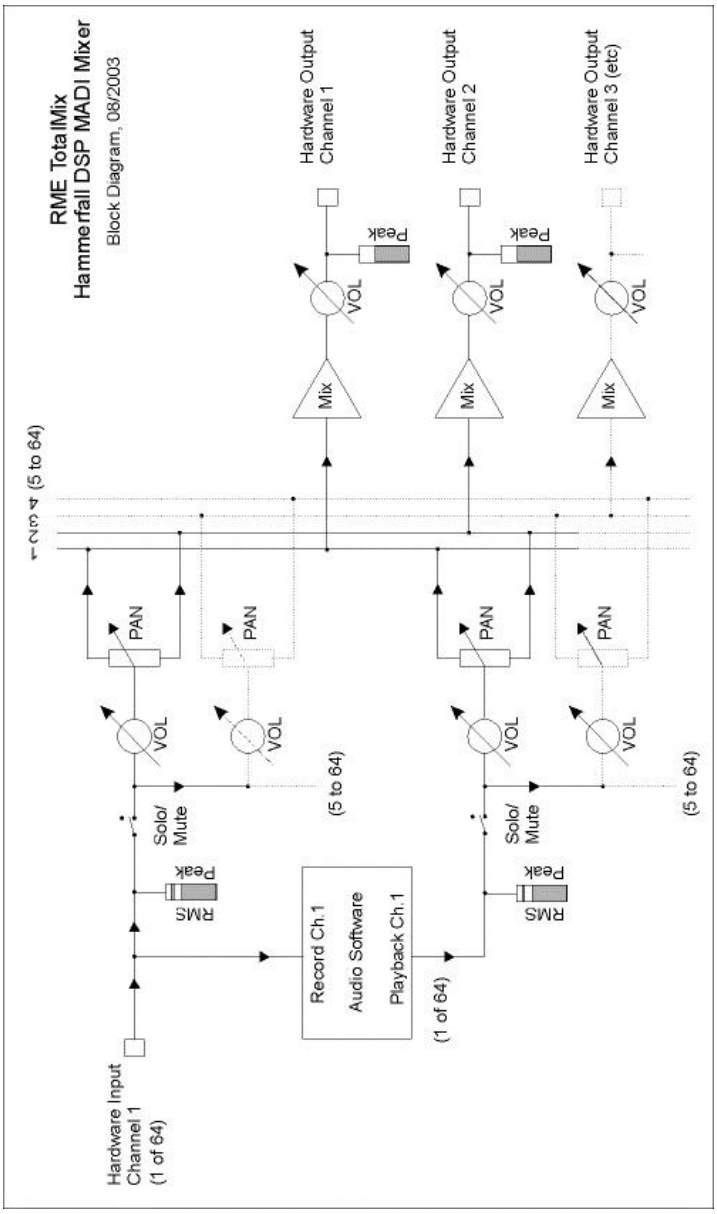
15. TotalMix: Routing and Monitoring

The Hammerfall DSP system includes a powerful digital real-time mixer. RME's unique TotalMix technology allows for nearly unlimited mixing and routing with all inputs and playback channels simultaneously.

Here are some typical applications for TotalMix:

- setting up delay-free submixes (headphone mixes)
- unlimited routing of inputs and outputs (free utilisation, patchbay function)
- distributing signals to several outputs at a time
- simultaneous playback of different programs over only one stereo channel
- mixing of the input signal to the playback signal (complete ASIO Direct Monitoring)
- integration of external devices (effects etc). in real-time

The block diagram of the TotalMix mixer of the HDSP MADI shows that the record signal always stays un-altered, but can be passed on as often as desired, even with different levels. The level meter of inputs and playback channels are connected pre-fader (due to the enormous routing capabilities). The level meters of the hardware's outputs are connected post-fader.



15.1 Elements of the Surface

The visible design of the TotalMix mixer is mainly determined by the architecture of the HDSP system:

- Upper row: hardware inputs. The level shown is that of the input signal, i. e. Fader independent. Per fader and routing window, any input channel can be routed and mixed to any hardware output (third row).
- Middle row: playback channels (playback tracks of the software). Per fader and routing window, any playback channel can be routed and mixed to any hardware output (third row).
- Lower row: hardware outputs. Because they refer to the output of a subgroup, the level can only be attenuated here (in order to avoid overloads), routing is not possible. This row has two additional channels, the analog outputs.

Every single channel has various elements:

Input and playback channels each have a mute and solo button.

Below each there is the panpot, realized as indicator bar (L/R) in order to save space.

In the window below this, the present level is displayed in RMS or Peak, being updated about every half a second. Overs are indicated here by an additional red dot.

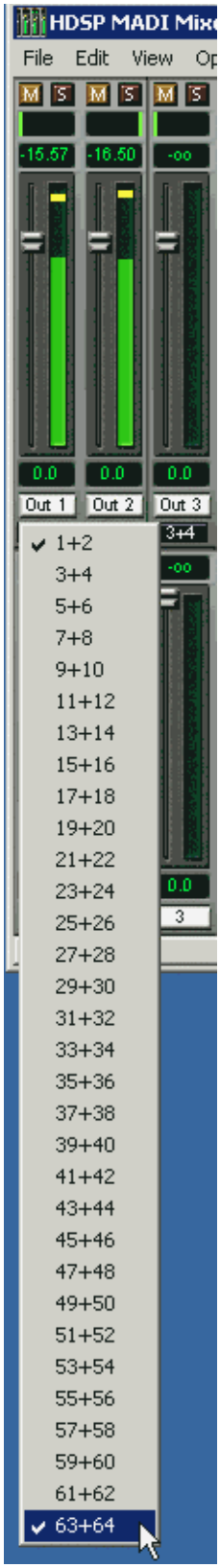
Then comes the fader with a levelmeter. The meter shows both peak values (zero attack, 1 sample is enough for displaying full scale) by means of a yellow line and mathematically correct RMS values by means of a green bar. The RMS display has a relatively slow time constant, so that it shows the average loudness quite well.

Below the fader, the current gain and panorama values are shown.

The white area shows the channel name, the black area shows the current routing target. Selecting one or more channels is done by clicking on the white label which turns yellow then.

15.2 Tour de TotalMix

In the following chapters we will explain all functions of the mixer surface step by step. Starting up TotalMix, the last settings are recalled automatically. When executing the application for the first time, a default file is loaded, sending all playback tracks 1:1 to the corresponding hardware outputs with 0 dB gain. The faders in the upper row are set to maximum attenuation (called m.a. in the following), so there is no monitoring of the input channels.



We will now create a submix on outputs 63/64 (headphones). Please start a multitrack playback and connect a headphone to the card. In playback channel 1 (labeled 'Out 1'), click onto the routing window below the label. A list pops up, showing a checkmark in front of '1+2'. Click onto '63/64'. The list disappears, the routing window no longer shows '1+2', but '63/64'. Now move the fader with the mouse. As soon as the fader value is unequal m.a., the present state is being stored and routing is activated. Move the fader button to around 0 dB. The present gain value is displayed below the fader in green letters. In the lower row, on channels 63 and 64, you can also see the level of what you are hearing. The level meter of the hardware output shows the outgoing level. Click into the area above the fader and drag the mouse in order to set the panorama, in this case the routing between channels 63 and 64. The present pan value is also being displayed below the fader.

Please carry out the same steps for 'Out 2' now, in order to route it to outputs 63/64 as well.

Often signals are stereo, i. e. a pair of two channels. It is therefore helpful to be able to make the routing settings for two channels at once. Press the Ctrl-key and click into the routing window of 'Out 3' with the key pressed. The routing list pops up with a checkmark at '3+4'. Click onto '63/64'. Now, channel 4 has already been set to '63/64' as well.

When you want to set the fader to exactly 0 dB, this can be difficult, depending on the mouse configuration. Move the fader close to the 0 position and now press the Shift-key. This activates the fine-mode, which stretches the mouse movements by a factor of 8. In this mode, a gain setting accurate to 0.1 dB is no problem at all.

Please set 'Out 4' to a gain of around -20 dB and the pan close to center. Now click onto the routing window. You'll now see two checkmarks, one at '3+4', the other one at '63/64'. Click onto '61/62'. The window disappears, fader and panpot jump to their initial values, the signal can now be routed to these channels. You can continue, until all entries have got a checkmark, i. e. you can send the signal to all outputs simultaneously.

You will certainly have noticed that the mix has not changed, while you were routing the channel to other outputs and setting different gain values. With all analog and most digital mixing desks, the fader setting would affect the level for every routed bus - not so for TotalMix. TotalMix allows for setting all fader values individually. Therefore the faders and the panpots jump to the appropriate setting as soon as another routing is chosen.

The checkmarks are un-checked by moving the fader to m.a. This setting deactivates the routing...why route if there is no level? Click onto '63/64' in the routing window, pull the fader down, open the routing window again - the checkmark is gone.

15.3 Submix View

Such a wide range of possibilities make it difficult to maintain the overview. Because practically all hardware outputs can be used for different submixes, as shown (up to 32 completely independent stereo submixes, 16 4-channel submixes etc.). And when opening the routing windows you might see an army of checkmarks, but you don't get an overwiev, i.e., how the signals come together and where. This problem is removed by the view mode 'Submix'. In this mode, all routing windows jump to the routing pair just being selected. You can then see immediately, which channels, which fader and pan settings make a submix (for example '63/64').

At the same time the Submix View simplifies setting up the mixer, as all channels can be set simultaneously to the same routing destination with just one click.

15.4 Mute and Solo

Mute works pre-fader, thus mutes all active routings of the channel. As soon as any Mute button is pressed, the Master Mute button lights up in the quick access area. It can switch all selected mutes off and on again. You can comfortably make mute groups to activate and deactivate this way.

The same holds true for the Solo and the Master Solo buttons. Solo is working as a solo-in-place. As soon as one Solo button is pressed, all other Mute buttons are activated and light up.

15.5 Hotkeys

TotalMix knows only a few, but very effective key combinations, that make setting the mixer up considerably easier and faster. The Shift-key for the fine-mode for faders and panpots has already been mentioned. But the Ctrl-key can do far more than changing the routing pairwise:

- Clicking anywhere into the fader area with the Ctrl-key pressed, sets the fader to 0 dB, -6 dB for the hardware outputs.
- Clicking anywhere into the pan area with the Ctrl-key pressed, sets the panorama to <C> meaning 'Center'.
- Clicking a Preset button while holding down Ctrl, the original (factory) preset will be loaded.
- Using Ctrl and any number between 1 and 8 (not on the numeric keypad!) will load the corresponding factory default preset
- Clicking the Card 2 button while holding down Ctrl opens a second TotalMix window.

The faders can also be moved pairwise, corresponding to the stereo-routing settings. This can be achieved by pressing the Alt-key and is especially comfortable when setting the SPDIF and analogue output level. Even the Panoramas can be operated with Alt, from stereo through mono to inversed channels. But also the Mute and Solo buttons (ganged or inversed switching!).

At the same time, TotalMix also supports combinations of these keys. If you press Ctrl and Alt at the same time, clicking with the mouse makes the faders jump to 0 dB pairwise, and they can be set pairwise by Shift-Alt in fine-mode.

Also very useful: the faders have two mouse areas. The first area is the fader button, which can be grabbed at any place without changing the position. This avoids unwanted changes when clicking onto it. The second area is the whole fader setting area. Clicking into this area makes the fader jump to the mouse at once. If you want to set several faders to m.a. for instance, it is sufficient to click onto the lower end of the fader path. Which happens pairwise with the Alt-key pressed.

Using the hotkeys I, O and P the complete row each of Input, Playback and Output channels can be toggled between visible and invisible. Hotkey S switches Submix view on/off. Those four hotkeys have the same functionality as the buttons in the View section of the Quick Access Panel. The Level Meter Setup dialog can be opened via F2 (as in DIGICheck and the Meter Bridge).

Hotkey M toggles Master Mute on/off (and with this performs a global mute on/off). Hotkey X toggles the Matrix view on/off (see chapter 16), hotkey T the mixer view.

Further hotkeys are available to control the configuration of the Level Meter (see chapter 15.10):

Key 4 or 6: Display range 40 or 60 dB
Key E or R: Numerical display showing Peak or RMS
Key 0 or 3: RMS display absolute or relative to 0 dBFS

15.6 The Quick Access Panel

This section includes additional options, further improving the handling of TotalMix. The Master button for Mute and Solo has already been described, they allow for group-based working with these functions.

In the **View** section the single rows can be made visible or invisible. If the inputs are not needed for a pristine playback mix, the whole upper row falls out of the picture after a click on the input button. If the hardware outputs don't interest you either, the surface can thus be reduced to the playback channels to save space. All combinations are possible.

Submix sets all routing windows to the same selection as described before. Deactivating Submix automatically recalls the previous view.

The mixer can also be made smaller horizontally and vertically. This way TotalMix can be made substantially smaller and space-saving on the desktop/screen, if you have to monitor or set only a few channels or level meters.

The **Presets** are one of the mightiest and most useful features of TotalMix. Behind the eight buttons, eight files are hidden (see next chapter). These contain the complete mixer state. Just try it: all faders and other settings follow the changing of preset(s) in real-time, just by a single mouse click. The Save button allows for storing the present settings in the present preset. You can change back and forth between a signal distribution, complete input monitoring, a stereo and mono mix, and various submixes without any problem.

Also here, RME's love for details can be seen. If any parameter is being altered after loading a preset (e. g. moving a fader), the preset display flashes in order to announce that something was changed, still showing, which state the present mix is based on.

If no preset button is lit, another preset had been loaded via the File menu and 'Open file'. Mixer settings can of course be saved the usual way, and with long file names.

Up to three Hammerfall DSP MADI can be used simultaneously. The **Card** buttons switch between the cards. Holding down Ctrl while clicking on button Card2 will open a second window, instead of replacing the current window content.

The number of channels is reduced automatically when entering Double or Quad Speed mode (96 / 192 kHz). The display is adjusted accordingly, and all fader settings remain stored (even the invisible ones).

15.7 Presets

TotalMix includes 8 factory presets, stored within the program. But the presets can be changed at any time, because TotalMix stores and reads the changed presets form the files **preset11.mix** to **preset81.mix**. These files are found in the hidden directory *Documents and Settings, <Username>, Local Settings, Application Data, RME TotalMix.* The first number indicates the current preset, the second number the current card/system.

This method offers two major advantages:

- Presets modified by the user will not be overwritten when reinstalling or updating the driver
- The factory presets remain unchanged, and can be reloaded anytime.

The original factory presets can be reloaded by holding down the Ctrl-key and clicking on any preset button.

Using 64, 88.2 and 96 kHz, TotalMix loads special factory presets with a submix on channels 31/32, for a working monitoring in these modes.

The 8 factory presets offer not only a useful functionality for TotalMix, but also a pretty good base to modify them to your personal needs.

Preset1.mix
All channels routed 1:1.

Details: All inputs maximum attenuation (m.a.). All playback channels 0 dB, routet to the same output. All outputs 0 dB. All channels prepared for all routings to left/right panning. Level display RMS +3 dB not activated.

Note: This preset is *Default*, offering the standard functionality of a I/O-card.

Preset2.mix
Details: All channels routed 1:1, playback monitoring plus input monitoring 1:1.

Preset3.mix
Details: All channels routed 1:1. No playback and no input monitoring. All faders down.

Preset4.mix
Details: All channels routed 1:1. As preset 2, but all inputs muted.

Note: This preset is default for ZLM and MME Mix/Replace monitoring. The factory preset 4 will also be loaded by a click on *Load Def*.

Preset5.mix
Details: All channels routed 1:1, playback monitoring. Submix of all playback channels to channel 63/64 (headphone monitoring). Hardware output 63/64 selected and faders at –12 dB.

Preset6.mix
Details: as preset 5, but submix of all input channels to channels 63/64 (headphone monitoring.

Preset7.mix
Details: as preset 5, but submix of all input and playback channels to channels 63/64 (headphone monitoring). View Submix active.

Preset8.mix
Description: Panic. As Preset 4, but also all playback channels muted (no output signal)

15.8 Monitor

The Monitor section of the Quick Access Panel is only valid for our Windows MME driver, i.e. when using programs like WaveLab, Soundforge, Sonar or Samplitude.

Monitor offers two advanced automated monitoring solutions. Monitoring will be controlled either by special commands directly from the recording software (Samplitude/Sequoia/SAWStudio, mode ZLM), or by the recording state itself (mode Mix/Replace).

The basic method used is as simple as it is clever: ZLM and record are controlling the Mute buttons of the corresponding channels. For this to work, Mute must be activated on the record's Input channel. The fader must not be set to m.a..

A click on **Load Def.** will load a template (factory preset 4), which can be used to verify and test this functionality. But it is also possible to use any other mixer state, as long as the recording channels are muted manually.

As soon as a recording starts, the corresponding channels will be unmuted, i.e. the input signal will be processed according to the current mixer settings. TotalMix lets you check the whole process visually, as the Mute buttons in TotalMix will be switched on and off automatically.

In **Mix** mode the input signal will be mixed on the outputs when record is active. In **Replace** mode the Mute button of the corresponding playback channel will be activated, so that the input signal replaces the playback signal.

ZLM is a special function for tape machine style monitoring when doing punch-ins and outs. For this to work the option 'Hardware monitoring during punch' has to be activated in Samplitude/Sequoia. Then at each punch-in the corres-ponding Mute buttons will be deactivated, at punch-out they will be reactivated.

All settings can be changed and configured in real-time.

15.9 Menu Options

Always on Top: When active (checked) the window of TotalMix will always be on top of the Windows desktop.
Note: This function may result in problems with windows containing help text, as the TotalMix window will even be on top of those windows, so the help text isn't readable.

Deactivate Screensaver: When active (checked) any activated Windows screensaver will be disabled temporarily.

Ignore Position: When active, the windows size and position stored in a file or prest will not be used. The routing will be activated, but the window will not change

ASIO Direct Monitoring: When de-activated any ADM commands will be ignored by TotalMix. In other words, ASIO Direct Monitoring is globally de-activated.

15.10 Level Meter

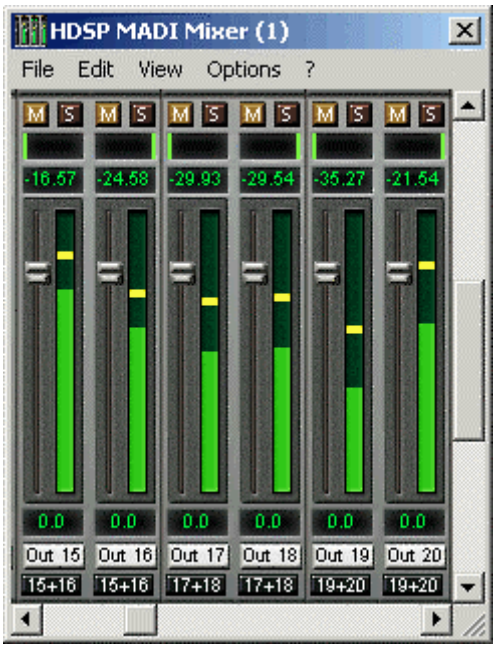
Having set a new standard with the level meters of DIGICheck, Hammerfall DSP goes even further: The calculation of the Peak, RMS and Over is realized in hardware, in order to be capable of using them independent of the software in use, and to significantly reduce the CPU load.

The level meters integrated in TotalMix - considering their size - cannot be compared with the *HDSP Meter Bridge* (chapter 21.2). Nevertheless they already include many useful functions.

Peak and RMS is displayed for every channel. 'Level Meter Setup' (Menu Options or F2) or direct keyboard entry (*hotkeys*) makes various options available:

- Display range 40 or 60 dB (*hotkey 4 or 6*)
- Release time of the Peak display (Fast/Medium/Slow)
- Numerical display selectable either Peak or RMS (*Hotkey E or R*)
- Number of consecutive samples for Overload display (1 to 15)
- RMS display absolute or relative to 0 dBFS (*Hotkey 3 or 0*)

The latter is a point often overlooked, but nonetheless important. RMS shows 3 dB less for sine signals. This is mathematically correct, but not very reasonable for a level meter. Therefore, we had corrected DIGICheck's RMS display by 3 dB, a full scale sine signal shows both 0 dBFS Peak and RMS. This setting also yields directly readable signal-to-noise values, while other applications (like WaveLab) will show a value 3 dB better than actual (because the reference is not 0 dB, but -3 dB).



The value displayed in the text field is independent of the setting 40/60 dB, it represents the full 24 bit range of the RMS measurement, thus making possible a SNR measurement 'RMS unweighted', which you would otherwise need extremely expensive measurement devices for. An ADI-8 DS connected to an ADI-648 and then to the MADI card will therefore show around -113 dBFS on all 8 channels.

This level display will constantly bring the reduced dynamic range of your equipment, maybe of the whole studio, in front of your eyes. Nice to have everything 24 bit - but still noise and hum everywhere in the range around -90 dB or worse... sorry, but this is hard reality. The up-side about it is that TotalMix allows for constantly monitoring the signal quality without effort. Thus it can be a valuable tool for sound optimization and error removal in the studio.

Measuring SNR (Signal to Noise) requires to press R (for RMS) and 0 (for referring to 0 dBFS, a full scale signal). The text display will then show the same value as an expensive measurement system, when measuring 'RMS unweighted'.

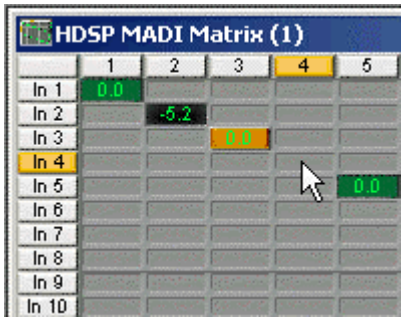
16. The Matrix

The mixer window of TotalMix looks and operates similar to mixing desks, as it is based on a conventional stereo design. The matrix display presents a different method of assigning and routing channels, based on a single channel or monaural design. The matrix view of the HDSP mixer looks a works like a conventional patchbay, adding functionality way beyond comparable hardware and software soutions. While most patchbays will allow you to connect inputs to outputs with just the original level (1:1, or 0 dB, as known from mechanical patchbays), TotalMix allows you to use a freely definable gain value per crosspoint.

Matrix and TotalMix are different ways of displaying the same processes. Because of this both views are always fully synchronized. Each change in one view is immediately reflected in the other view as well.

16.1 Elements of the Surface

The visible design of the TotalMix Matrix is mainly determined by the architecture of the HDSP system:



- Horizontal labels: 64 hardware outputs
- Vertical labels: 64 hardware inputs. Below 64 playback channels (software playback)
- Green 0.0 dB field. Standard 1:1 routing
- Black gain field. Shows the current gain value as dB
- Orange gain field. This routing is muted.

To maintain overview when the window size has been reduced, the left and upper labels are floating. They won't left the visible area when scrolling.

16.2 Operation

Using the Matrix is a breeze. It is very easy to indentify the current crosspoint, because the labels light up in orange according to the mouse position.

If input 1 is to be routed to output 1, use the mouse and click one time on crosspoint **In1** / **1**. The green 0.0 dB field pops in, another click removes it. To change the gain (equals the use of a different fader position, see simultaneous display of the mixer view), hold Ctrl down and drag the mouse up or down, starting from the gain field. The value within the field changes accordingly. The corresponding fader in the mixer view is moving simultaneously, in case the currently modified routing is visible.

Note the difference between the left side, representing the inputs and software playback sources, and the upper side, representing the hardware outputs. If you move a fader in row 1 or 2 in TotalMix, only the specific levels (max. 2) of this routing will change in the Matrix. But moving a fader in row 3 will make all vertically activated levels move at once (for example 63/64, analog output).

A gain field marked orange indicates activated Mute status. Mute can only be changed in the mixer view.

16.3 Advantages of the Matrix

The Matrix not always replaces the mixer view, but it significantly enhances the routing capabilities and - more important - is a brilliant way to get a fast overview on all active routings. It shows you in a glance what's going on. And since the Matrix operates monaural, it is very easy to set up specific routings with specific gains.

Example 1: You want TotalMix to route all software outputs to all corresponding hardware outputs, and have a submix of all software outputs on the analog output (equals factory preset 5). Setting up such a submix is fast and easy. But how to check at a later time, that all settings are still exactly the way you wanted them to be?

So far the only way to check that TotalMix is correctly set up this way, is to activate Submix view, step through all existing software outputs, and have a very concentrated look at the faders and displayed levels of each routing. That doesn't sound comfortably nor error-free, right? Here is where the Matrix shines. In the Matrix view, you simply see a line from upper left to lower right, all fields marked as unity gain. Plus two rows vertically all at the same level setting. You just need 2 seconds to be sure no unwanted routing is active anywhere, and that all levels match precisely!

Example 2: The Matrix allows you to set up routings which would be nearly impossible to achieve by fiddling around with level and pan. Let's say you want to send input 1 to output 1 at 0 dB, to output 2 at -3 dB, to output 3 at -6 dB and to output 4 at -9 dB. Each time you set up the right channel (2/4), the change in pan destroys the gain setting of the left channel (1/2). A real hassle! In Matrix view, you simply click on the corresponding routing point, set the level via Ctrl-mouse, and move on. You can see in the desk view how level and pan changes automatically when performing the second (fourth...) setting.

17. TotalMix Super-Features

17.1 ASIO Direct Monitoring

Start Samplitude, Sequoia, Cubase or Nuendo and TotalMix. Activate ADM (ASIO Direct Monitoring), and move a fader in the ASIO host. Now watch the corresponding fader in TotalMix magically move too. TotalMix reflects all ADM gain and pan changes in realtime. Please note that faders only move when the currently activated routing (currently visible routing) corresponds to the one in the ASIO host. Also note that the Matrix will show any change, as it shows all possible routings in one view.

With this TotalMix has become a wonderful debugging tool for ADM. Just move the host's fader and pan, and see what kind of ADM commands TotalMix receives.

The hardware output row faders are included in all gain calculation, in every possible way. Example: you have lowered the output level of a submix, or just a specific channel, by some dB. The audio signal passed through via ADM will be attenuated by the value set in the third row.

17.2 Selection and Group-based Operation

Click on the white name label of channel 1 and 2 in TotalMix. Be sure to have channel 3's fader set to a different position and click on its label too. All three labels have changed to the colour orange, which means they are selected. Now moving any of these faders will make the other faders move too. This is called 'building a group of faders', or ganging faders, maintaining their relative position.

Building groups or ganging can be done in any row, but is limited to operate horizontally within one row. If you usually don't need this, you can at least gang the analog outputs. The advantage over holding the Alt-key is that Alt sets both channels to the same level (can be handy too), while grouping via selection will retain any offset (if you need one channel to be louder all the time etc.).

Note: if you move the mouse so that any channel reaches upper or lower maximum position, and release the mouse button, the relative position is lost.

Tip: gang some submixes and watch all routing levels change like crazy in the Matrix view.

17.3 Copy Routings to other Channels

TotalMix allows to copy complete routing schemes of inputs and outputs.

Example 1: You have input 5 (guitar) routed within several submixes/hardware outputs (= headphones). Now you'll get another input with keyboards that should appear in the same way on all headphones. Select input 5, open the menu *Edit*. It shows 'Copy In 5'. Now select the desired new input, for example In 8. The menu now shows 'Paste In 5 to In 8'. Click on it - done. If you are familiar with this functionality just use Ctrl-C and Ctrl-V. Else the self updating menu will always let you know what actually will happen.

Tip: have the Matrix view open when doing this. It will show the new routings immediately, so copying is easier to understand and to follow.

Example 2: You have built a comprehensive submix on outputs 4/5, but now need the exact same signal also on the outputs 6/7. Click on Out 4, Ctrl-C, click on Out 6, Ctrl-V, same with 5/7 - you're done!

The Matrix shows you the difference between both examples. Example 1 means copying lines (horizontally), while example 2 means copying rows (vertically).

Example 3: Let's say the guitarist finished his recording, and you now need the same signal again on all headphones, but this time it comes from the recording software (playback row). No problem, you can even copy between rows 1 and 2 (copying between row 3 and 1/2 isn't possible).

But how to select while a group is active? De-selecting the group first? Not necessary! TotalMix always updates the copy and paste process with the *last* selection. This way you don't have to de-activate any group-selections when desiring to perform a copy and paste action.

17.4 Delete Routings

The fastest way to delete complex routings: select a channel in the mixer view, click on the menu entry *Edit* and select *Delete*. Or simply hit the Del-key. Attention: there is no undo in TotalMix, so be careful with this function!

18. MADI Basics

MADI, the serial **M**ultichannel **A**udio **D**igital **I**nterface, has been defined already in 1989 as an extension of the existing AES3 standard following several manufacturers' wish. The format also known as AES/EBU, a balanced bi-phase signal, is limited to two channels. Simply put, MADI contains 28 of those AES/EBU signals in serial, i. e. after one another, and the sample rate can still even vary by +/-12.5%. The limit which cannot be exceeded is a data rate of 100Mbit/s.

Because an exact sampling frequency is used in most cases, the 64 channel mode was introduced officially in 2001. It allows for a maximum sample rate of 48 kHz + ca. 1%, corresponding to 32 channels at 96 kHz, without exceeding the maximum data rate of 100 Mbit/s. The effective data rate of the port is 125 Mbit/s due to additional coding.

Older devices understand and generate only the 56 channel format. Newer devices often work in the 64 channel format, but offer still no more than 56 audio channels. The rest is being eaten up by control commands for mixer settings etc.. The ADI-648 and the HDSP MADI show that this can be done in a much better way, with an invisible transmission of 16 MIDI channels and the MADI signal still being 100% compatible.

For the transmission of the MADI signal, proved methods known from network technology were applied. Most people know unbalanced (coaxial) cables with 75 Ohms BNC plugs, they are not expensive and easy to get. The optical interface is much more interesting due to its complete galvanic separation, but for many users it is a mystery, because very few have ever dealt with huge cabinets full of professional network technology. Therefore here are some explanations regarding 'MADI optical'.

- The cables used are standard in computer network technology. They are thus not at all expensive, but unfortunately not available in every computer store.

- The cables have an internal fibre of only 50 or 62.5 µm diameter and a coating of 125 µm. They are called network cables 62.5/125 or 50/125, the former mostly being blue and the latter mostly being orange. In most cases, they are not (!) glass fibre cables, but plastic fibre cables (POF, plastic optical fibre).

- The plugs used are also an industry standard and called SC. Please don't mix them up with ST connectors, which look similar to BNC connectors and are being screwed. Plugs used in the past (MIC/R) were unnecessarily big and are not being used any longer.

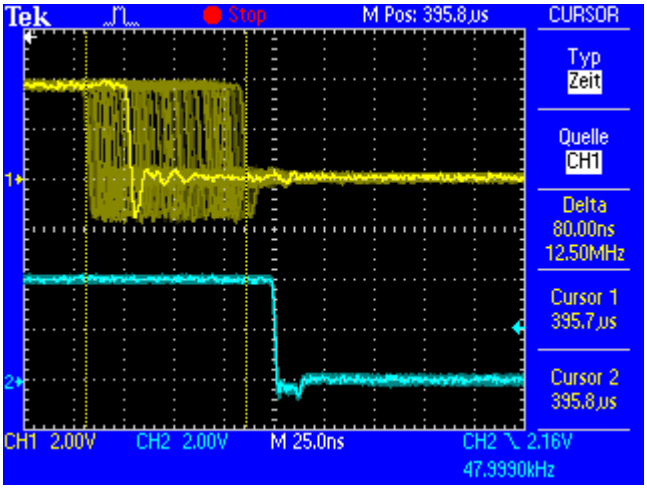
- The cables are available as a duplex variant (2 cables being glued together) or as a simplex variant (1 cable). The HDSP MADI's opto module supports both variants.

- The transmission uses the multimode technique which supports cable lengths of up to almost 2 km. Single mode allows for much longer distances, but it uses a completely different fibre (8 µm). By the way, due to the wave-length of the light being used (1300 nm), the optical signal is invisible to the human eye.

19. SteadyClock

The Hammerfall DSP MADI's SteadyClock technology guarantees an excellent performance in all clock modes. Its highly efficient jitter suppression refreshes and cleans up any clock signal, and provides it as reference clock at the word clock output.

SteadyClock has been originally developed to gain a stable and clean clock from the heavily jittery MADI data signal. The embedded MADI clock suffers from about 80 ns jitter, caused by the time resolution of 125 MHz within the format. But common jitter values for other devices are 5 ns, while a very good clock will have less than 2 ns.



The picture to the right shows the MADI input signal with 80 ns of jitter (top graph, yellow). Thanks to SteadyClock this signal turns into a clock with less than 2 ns jitter (lower graph, blue). The cleaned and jitter-freed signal can be used as reference clock for any application, without any problem. The signal processed by SteadyClock is of course not only used internally, but also available at the card's word clock output.

20. Hotline - Troubleshooting

20.1 General

The newest information can always be found on our website www.rme-audio.com, section FAQ, Latest Additions.

The input signal cannot be monitored in real-time

- ASIO Direct Monitoring has not been enabled, and/or monitoring has been disabled globally.

Playback works, but record doesn't:

- Check that there is a valid signal at the input. If so, the current sample frequency is displayed in the Settings dialog.
- Check whether the Hammerfall DSP has been selected as recording device in the audio application.
- Check whether the sample frequency set in the audio application ('Recording properties' or similar) matches the input signal.
- Check that cables/devices have not been connected in a closed loop. If so, set the systems's clock mode to 'Master'.

Crackle during record or playback:

- Increase the number and size of buffers in the ‘Settings’ dialog or in the application.
- Try different cables (coaxial or optical) to rule out any defects here.
- Check that cables/devices have not been connected in a closed loop. If so, set the system’s clock mode to ‘Master’.
- Increase the buffer size of the hard disk cache.
- Activate Busmaster mode for the hard disks.
- In case of a recently done BIOS update of the motherboard: Propably 'Load BIOS Defaults' was loaded instead of 'Load Setup Defaults'. This sets the 'PCI Latency Timer' to 0 (default: 32).

Low Latency ASIO operation under Windows 2000/XP on single CPU systems:

- To use ASIO at lowest latencies under Windows 2000/XP even when only having one CPU, the system performance has to be optimized for background tasks. Go to Control Panel/System/Advanced/Performance Options. Change the default 'Applications' to 'Background tasks'. The lowest usable latency will drop from 23 ms to around 3 ms. This is no issue when using dual CPU systems.

20.2 Installation

More information on installation problems (which fortunately are very seldom, thanks to Plug and Play), can be found in the Tech Info 'Installation problems'. It is located in the directory **\rmeaudio.web\techinfo** on the RME Driver CD.

HDSP MADI is normally found in the Device Manager (>Settings/Control Panel/System<), category 'Sound-, Video- and Gamecontroller'. A double click on ' Hammerfall DSP MADI' starts the properties dialog. Choosing 'Resources' shows Interrupt and Memory Range.

The newest information on hardware problems can always be found on our website www.rme-audio.com, section FAQ, Hardware Alert: about incompatible hardware.

The dialog 'New hardware component found’ does not appear:

- Check whether the PCI interface is correctly inserted in the PCI slot.

The card and drivers have been installed correctly, but playback does not work:

- Check whether the Hammerfall DSP appears in the Device Manager. If the ' Hammerfall DSP’ device has a yellow exclamation mark, then there is an address or interrupt conflict.
- Even if there is no yellow exclamation mark, it is worth checking the ‘Resources’ tab anyway.
- Check whether the Hammerfall DSP has been selected as current ASIO device.

21. HDSP Software

21.1 DIGICheck 4.0

The DIGICheck software is a unique utility developed for testing, measuring and analysing digital audio streams. Although the DIGICheck software is fairly self-explanatory, it still includes a comprehensive online help. DIGICheck 4.0 operates as multi-client ASIO host, and can therefore be used in parallel to any software, be it MME, ASIO or GSIF, both inputs and even outputs (!). The following is a short summary of the available functions:

- **Level Meter**. High precision 24-bit resolution, 2/8/26 channels. Application examples: Peak level measurement, RMS level measurement, over-detection, phase correlation measurement, dynamic range and signal-to-noise ratios, RMS to peak difference (loudness), long term peak measurement, input check. Oversampling mode for levels higher than 0 dBFS.
- **Vector Audio Scope**. World wide unique Goniometer showing the typical afterglow of a oscilloscope-tube. Includes Correlation meter and level meter.
- **Spectral Analyser.** World wide unique 10-, 20- or 30-band display in analog bandpass-filter technology. 192 kHz-capable!
- **Bit Statistics & Noise**. Shows the true resolution of audio signals as well as errors and DC offset. Includes Signal to Noise measurement in dB and dBA, plus DC measurement.
- **Totalyser**. Spectral Analyser and Vector Audio Scope as one window function.
- **Channel Status Display**. Detailled analyzis and display of SPDIF and AES/EBU Channel Status data.
- **Completely multi-client.** Open as many measurement windows as you like, on any channels and inputs or outputs!

To install DIGICheck, go to the **\DIGICheck** directory on the RME Driver CD and run *setup.exe*. Follow the instructions prompted on the screen.

21.2 HDSP Meter Bridge 2.0

The Hammerfall DSP Meter Bridge is a unique, highly flexible and handy tool for level metering. As opposed to DIGICheck, the HDSP Meter Bridge receives all level data directly from the hardware. Thus the Meter Bridge can not only run in parallel to any other program, but as Peak, Over and RMS calculations are performed directly in hardware, the CPU load caused is limited to the graphics routines – and is near zero on todays computers! Available functions are:

- **2-channel Level Meter**. High precision 24-bit resolution. Application examples: Peak level measurement, RMS level measurement, over-detection, dynamic range and signal-to-noise ratios, RMS to peak difference (loudness), long term peak measurement, input check.
- **8-channel Level Meter**. As 2-channel Level Meter, but combined Peak/RMS display.
- **Global Level Meter**. As 8-channel Level Meter, but displays all channels of a card.
- **Completely multi-client.** Open as many measurement windows as you like, on any channels and inputs or outputs!

Although the HDSP Meter Bridge is fairly self-explanatory, it still includes a comprehensive online help. To start press F2 and F3, the most important hotkeys.

To install the HDSP Meter Bridge, go to the **\HDSP Meter Bridge** directory on the RME Driver CD and run *setup.exe*. Follow the instructions prompted on the screen.

22. Accessories

Parts of the HDSP MADI are available separately.

Part Number	Description
BOHDSP9652	MIDI breakout cable
VKHDSP9652	14-conductor flat ribbon cable

23. TECH INFO

Not all information to and around our products fit in a manual. Therefore RME offers a lot more and detailed information in the **Tech Infos**. The very latest Tech Infos can be found on our website, section News & Infos, or the directory **\rmeaudio.web\techinfo** on the RME Driver CD. These are some of the currently available *Tech Infos*:

Synchronization II (DIGI96 series)
Digital audio synchronization - technical background and pitfalls.

Installation problems
Problem descriptions and solutions.

Information on driver updates
Lists all changes in the drivers.

Configuring Logic, Samplitude, Cubase, Cakewalk, Sonar and SAWPlus32
Step by step instructions for use with RME cards.

DIGICheck: Analysis, tests and measurements with the DIGI96 series
A description of DIGICheck, including technical basics.

ADI-8 Inside
Technical information about the RME ADI-8 (24-bit AD/DA converter).

HDSP System: Notebook Basics - Notebook Hardware
HDSP System: Notebook Basics - The Audio Notebook in Practice
HDSP System: Notebook Basics - Background Knowledge and Tuning
HDSP System: Notebook Tests - Compatibility and Performance
Many background information on laptops. Tests of notebooks

HDSP System: TotalMix - Hardware and Technology
HDSP System: TotalMix - Software, features, operation
The digital mixer of the Hammerfall DSP in theory and practise

24. Warranty

Each individual Hammerfall DSP undergoes comprehensive quality control and a complete test in a PC environment at RME before shipping. This may cause very slight signs of wear (if it looks like it was used one time before - it was). The usage of high grade components allows us to offer a full two year warranty. We accept a copy of the sales receipt as valid warranty legitimation.

RME's replacement service within this period is handled by the retailer. If you suspect that your card is faulty, please contact your local retailer. The warranty does not cover damage caused by improper installation or maltreatment - replacement or repair in such cases can only be carried out at the owner's expense.

RME does not accept claims for damages of any kind, especially consequential damage. Liability is limited to the value of the Hammerfall DSP. The general terms of business drawn up by Synthax Audio AG apply at all times.

25. Appendix

RME news, driver updates and further product information are available on our website:

http://www.rme-audio.com

If you prefer to read the information off-line, you can load a complete copy of the RME website from the RME Driver CD (in the **\rmeaudio.web** directory) into your browser.

Trademarks

All trademarks, registered or otherwise, are the property of their respective owners. RME, DIGI96, SyncAlign, ZLM, SyncCheck, DIGICheck and Hammerfall are registered trademarks of RME Intelligent Audio Solutions. TotalMix, Intelligent Clock Control and TMS are trademarks of RME Intelligent Audio Solutions. Alesis and ADAT are registered trademarks of Alesis Corp. ADAT optical is a trademark of Alesis Corp. Microsoft, Windows 98/SE/ME and Windows 2000/XP are registered trademarks or trademarks of Microsoft Corp. Steinberg, Cubase and VST are registered trademarks of Steinberg Media Technologies GmbH. ASIO is a trademark of Steinberg Media Technologies GmbH. Pentium is a registered trademark of Intel Corp.

Copyright © Matthias Carstens, 11/2003. Version 1.1
Current driver version: W2k: 2.0

Although the contents of this User's Guide have been thoroughly checked for errors, RME can not guarantee that it is correct throughout. RME does not accept responsibility for any misleading or incorrect information within this guide. Lending or copying any part of the guide or the RME Driver CD, or any commercial exploitation of these media without express written permission from RME Intelligent Audio Solutions is prohibited. RME reserves the right to change specifications at any time without notice.

26. CE and FCC Compliance Statements

CE

This device has been tested and found to comply with the EN55022 class B and EN50082-1 norms for digital devices, according to the European Council directive on counterpart laws in the member states relating to electromagnetic compatibility (EMVG).

FCC

This device has been tested and found to comply with the requirements listed in FCC Regulations, part 15 for Class 'B' digital devices. Compliance with these requirements provides a reasonable level of assurance that your use of this product in a residential environment will not result in harmful interference with other electronic devices.

This equipment generates radio frequencies and, if not installed and used according to the instructions in the User's Guide may cause interference harmful to the operation of other electronic devices.

Compliance with FCC regulations does not guarantee that interference will not occur in all installations. If this product is found to be the source of interference, which can be determined by turning the unit off and on again, please try to eliminate the problem by using one of the following measures:

- Relocate either this product or the device that is being affected by the interference
- Use power outlets on different branch circuits, or install AC line filters
- Contact your local retailer or any qualified radio and television engineer

When connecting external devices to this product, compliance to limits for a Class 'B' device requires the use of shielded cables.

FCC compliance statement: Tested to comply with FCC standards for home or office use.